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(54) Title: MULTICARRIER DIGITAL TRANSMISSION SYSTEM USING AN OQAM TRANSMULTIPLEXER
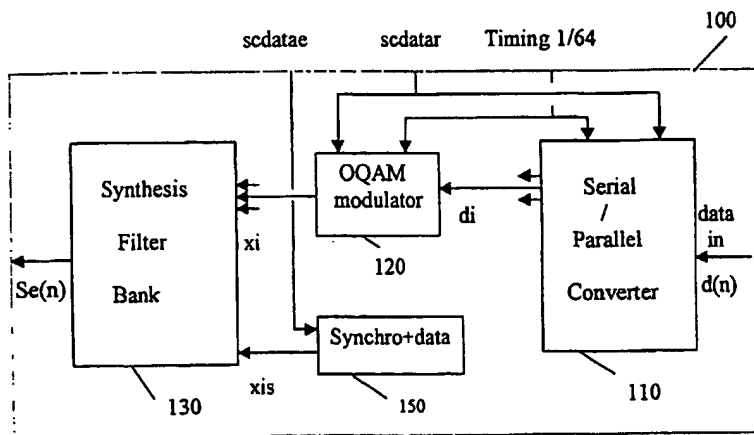


Digital Multicarrier Emitter

(57) Abstract: An emitter converts a serial input data stream into a set of parallel substreams. An OQAM modulator (120) receives and supplies each substream to an input of a synthesis filter bank (130). Synchronization input(s) of the filter bank receive an OQAM signal that carries a data sequence. This sequence contains frame, superframe and hyperframe synchronization patterns, and specifies the number of bits allocated to each subchannel. A receiver (200) includes an analysis filter bank (210) that decomposes the multicarrier signal into a set of elementary signals for each subchannel. A synchronizing processing block (270) receives output(s) of the filter bank for synchronization subchannel(s). The block (270) includes a first cascade of blocks that control receiver sampling times, and a second cascade of blocks that extract synchronization patterns and subchannel bit assignment data. Other filter bank outputs are each coupled to a cascade subchannel equalizer (220) followed by a data extractor (230) and a parallel-to-serial converter (240).

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## Multicarrier Digital Transmission System
## Using an OQAM Transmultiplexer

### Technical Field

5       The present invention relates to systems for the transmission of digital data over a communication channel using a multicarrier modulation and, more particularly, to an improved OQAM transmultiplexer method for use in such systems.

### Background Art

10       A multicarrier transmission system, as opposed to a single carrier system, uses a set of different frequencies distributed in the transmission channel frequency band to carry the data. The main advantage is that the bit rate can be adjusted for each carrier, according to the noise and distortion power in the

15  vicinity of this carrier. Thus, a better approximation of the theoretical information capacity limit can be expected and, particularly, poor quality channels can be exploited, like some wireless communication channels, power lines or the telephone subscriber lines at high or very high frequencies. A detailed

20  description of existing multicarrier transmission systems and their merits compared to single carrier systems is given in the book by W.Y.Chen: « DSL-Simulation Techniques and Standards Development for Digital Subscriber Line Systems », MacMillan

25  Technical Publishing, Indianapolis, USA, 1998.
       In order to efficiently perform multicarrier transmission, two basic approaches have been considered so far. The first one and most widely used is called OFDM (Orthogonal Frequency Division Multiplexing) or DMT (Digital Multi-Tone) and it is

30  based on the FFT (Fast Fourier Transform). It has been a subject of intense research and development efforts. In that scheme, the data are arranged in blocks which are transmitted by orthogonal carriers, and separated by guard times, which have to be greater than the channel impulse response, to preserve the orthogonality

35  of the carriers at the receiving side. In spite of the potential of the approach, OFDM/DMT suffers from a number of weaknesses, which, overall, make it perform hardly better than single carrier transmission: a complex time equalizer has to be introduced in

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front of the receiver to reduce the channel impulse response length, very precise time synchronization is necessary, a long initialization phase is required, and the carriers, and subchannels, are poorly separated, which reduces the capacity of 5 the system in the presence of jammers. In fact, a good quality channel is necessary for that scheme to work properly. Ample documentation can be found in the literature and a good list of references is given in the book by W.Y.Chen.

A second approach aims at overcoming some of the OFDM/DMT 10 limitations through the use of more sophisticated transforms than the FFT, namely, lapped transforms and wavelet transforms. The idea is to improve the separation between carriers, or subchannels. A lot of theoretical work has been done about this subject, see for example the paper by S.D.Sandberg and 15 M.A.Tzannes: « Overlapped Discrete Multitone Modulation for High Speed Copper Wire Communications », IEEE-JSAC, Vol.13, N°9, Dec. 1995. Although they improve the subchannel separation, the lapped and wavelet approaches still retain some of the crucial OFDM/DMT limitations and particularly the time synchronization 20 requirements.

In fact, the ideal approach for multicarrier transmission is one in which the subchannels are made independent and this is achieved by filter banks. This has been recognized a long time, and an efficient implementation of filter banks for transmission 25 systems, based on the combination of an FFT processor with a polyphase network, has been presented in the paper by M.Bellanger and J.Daguet: « TDM-FDM Transmultiplexer: Digital Polyphase and FFT », IEEE Transactions on Communications, Vol.COM-22, Sept.1974. Later on, a technique called OQAM (Orthogonal 30 Quadrature Amplitude Modulation) has been proposed for multicarrier transmission with filter banks, see the paper by B.Hirosaki, « An Orthogonally Multiplexed QAM System Using the Discrete Fourier Transform », IEEE Trans. on Communications, Vol.COM-29, July 1981. Its main feature is that the subchannel 35 sampling rate is twice the Nyquist subchannel frequency, or subchannel spacing, and the data are transmitted alternatively on the real and the imaginary part of the complex signal in any subchannel, with, again, an alternation between two adjacent

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subchannels. With this technique, intersymbol interference is
eliminated in a subchannel and between adjacent subchannels. The
distortions introduced by the transmission channel can be
eliminated by a multibranch equalizer in each subchannel.
5  Recently, it has been shown that a single branch equalizer can
be used, see the paper by L.Qin and M.Bellanger, « Equalization
issues in Multicarrier Transmission Using Filter Banks », Annals
of Telecommunications, Vol.52, N°1-2, January 1997.

In spite of its potential theoretical advantages, the OQAM
10  multicarrier approach is seldom considered for implementation in
practical systems.  A key reason is that the real/imaginary
alternation principle raises problems, which have not been
adequately solved thus far, for the equalization algorithms, the
carrier synchronization and the system timing organization.

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Disclosure of Invention
It is a primary object of the present invention, to achieve
a highly robust and efficient multicarrier transmission system,
using the transmultiplexer concept of filter banks in combination
20  with OQAM modulation.
The above and other objects are achieved in accordance with
the present invention wherein a signal containing synchronization
patterns which define a timing structure consisting of frames,
superframes and hyperframes is fed, in the emitter, to the input
25  of one or several subchannels reserved for synchronization, the
relevant information being carried by the magnitude or envelop
of the complex OQAM signal.  The same signal also contains
service data giving the number of bits allocated to each
subchannel.  In the other subchannels, a short fixed pattern is
30  introduced periodically to serve as a reference signal for the
subchannel equalizers in the receiver.
At the output of the analysis filter bank in the receiver,
the signal corresponding to the synchronization subchannel(s) is
coupled to a first cascade containing an amplitude equalizer, an
35  envelop detector and a filter that delivers the control signal
for the phase lock loop associated with the receiver clock
generator and to a second cascade containing an amplitude and
phase equalizer, a data detector and a block for the identifica-

tion of the superframe and hyperframe synchronization patterns and the service data extraction. The other outputs of the analysis filter bank are coupled to cascade subchannel equalizers consisting of three elements each, namely an amplitude equalizer, a phase equalizer and a fine equalizer. Every subchannel equalizer is followed by a data extractor and both use the information provided by the synchronization subchannel to complete their functions. The output error signals are used to determine the number of bits assigned to each subchannel and the information is transmitted to the distant terminal via the synchronization subchannel(s) every hyperframe.

With the system of the invention, no initialization specific sequence is necessary at the beginning of a transmission session or after an interruption, and the bit rate distribution among the subchannels can be adjusted continuously in time during the transmission.

These and other features, objects and advantages will be understood or apparent to those of ordinary skill in the art from the following detailed description of the preferred embodiment as illustrated in the various drawing figures.

Brief Description of Drawings

The invention will be more fully appreciated from the following detailed description when the same is considered in connection with the accompanying drawings, in which:

FIG. 1 is a simplified schematic block diagram of a multicarrier transmission system, in accordance with the present invention;

FIG. 2 is a schematic block diagram of the digital multicarrier emitter;

FIG. 3 is a drawing showing the waveform of the envelop of the signal in the synchronization subchannel;

FIG. 4 is a schematic block diagram of the digital multicarrier receiver;

FIG. 5 is a schematic block diagram of the subchannel equalizer; and

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FIG. 6 is a schematic block diagram showing the functions involved in the processing of the signals received in the synchronization channel.

5   Best Mode for Carrying Out the Invention

The block diagram of a multicarrier transmission system is shown in FIG. 1, for the case of a digital subscriber telephone line application. The input data stream d(n) is fed to a multicarrier emitter 100, which forms the signal Se(n) and is

10   itself coupled to a module 10 that performs the digital-to-analog (D/A) conversion and the emitter analog front end functions. Basically, the emitter analog front end consists of an amplifier and a low-pass or pass-band filter to limit the spectrum sent to the hybrid circuit 11. The hybrid is connected to the twisted

15   pair line 14 and its receiving port is connected to a module 12 that performs the receiver analog front end functions and the analog-to-digital (A/D) conversion. The receiver analog front end consists of a low-pass or band-pass filter to prevent aliasing and a variable gain amplifier. If symmetric, or full

20   duplex transmission is contemplated, the A/D converter is coupled to an echo canceller 13 that ensures an adequate level of separation between the two directions of transmission. The echo canceller produces the signal Sr(n) and it is coupled to a multicarrier receiver 200 that delivers the output data stream

25   d'(n). A detailed description of the analog front ends, hybrid circuits, A/D and D/A converters as well as echo cancellers is given in W.Y.Chen's book.

The present invention is concerned with the multicarrier emitter block 100 and receiver block 200 shown in greater details

30   in FIGs. 2 and 4 respectively.

Turning to FIG. 2, the input data d(n) are processed by a cascade of three blocks, namely a serial/parallel converter 110, an OQAM modulator 120 and a synthesis filter bank (SFB) 130, to produce the emitted digital multicarrier signal Se(n). Converse-

35   ly, as shown in FIG. 4, the received multicarrier digital signal Sr(n) is processed by a cascade of 4 blocks, namely an analysis filter bank (AFB) 210, a subchannel equalizer 220, a data extractor 230 and a parallel/serial converter 240, to produce the

output data sequence d'(n). In the absence of transmission errors, d'(n) and d(n) are identical, except for a delay.

The filter banks, SFB **130** and AFB **210** consist of an FFT processor, coupled to a polyphase network as described in the paper by M.Bellanger and J.Daguet. Denoting by fs the sampling frequency of the multicarrier signal and by N the size of the FFT, which is twice the number of real subchannels, the subchannel frequency spacing is fs/N and the FSB and AFB operate at the rate 2(fs/N). For example, in subscriber line transmission, the following values may be selected: fs = 2048 kHz; N = 512; 2(fs/N) = 8 kHz.

The specificity of the filter banks, SFB **130** and AFB **210** resides in the values of their coefficients, that are the same for both, or very close. The filter bank coefficients are computed from a prototype filter frequency response H(f) that is half-Nyquist in the pass-band and provides the maximum attenuation in the stop-band. Therefore, the cascade of the filter banks SFB and AFB exhibits a frequency response $H^2(f)$ that satisfies the first Nyquist criterion. It is advantageous to have $H^2(f)$ satisfy also the second Nyquist criterion, because intermediate signal samples take on well defined values. For example, if the data samples fed to the real or imaginary part of a subchannel are ±1, the intermediate signal samples at the output of the SFB-AFB cascade are { +1; 0; -1 }.

A possible choice for the prototype filter frequency response is an approximation of the following function

$$Ht(f) = \cos(\pi\, N\, f\, /\, 2\, fs) \qquad ; \qquad 0 \le |f| \le fs\, /\, N \qquad (I)$$

$$Ht(f) = 0 \qquad\qquad ; \qquad fs\, /\, N \le |f| \le fs\, /\, 2$$

using the Fourier series expansion. Accordingly, a prototype filter with M=2P+1 coefficients has the following coefficient values, in the above example.

$$h_i = \frac{\cos(\pi\, i\, /\, 256)}{1 - (\, i\, /\, 128)^2} \qquad ; \qquad -P \le i \le P \qquad (2)$$

The input signal xi to the SFB block **130** that corresponds to subchannel i, is supplied by the OQAM modulator **120**, a device that associates the input data di to quantized signal samples xi according to predetermined rules, as is well known in data transmission and described for example in the book by W.Y.Chen.

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The specificity here is that the signal samples take on real and imaginary values alternatively to obey the OQAM principle and the number of levels is determined by an external control signal denoted « scdatar » in FIG. 2. The control signal scdatar

5   adjusts the number of bits transmitted by a subchannel to its estimated capacity, as will be explained below. For example, if 1 bit can be carried by subchannel i, the sample xi may take on the following values: ± 1. But, if 2 bits can be carried, then the sample xi may take on the values: -1.5; -0.5; +0.5; +1.5.

10      The serial-to-parallel converter **110** splits the input bit stream d(n) into as many substreams as used subchannels and, for each substream, constitutes groups of bits di, under the control of the external signal scdatar, to feed the OQAM modulator **120**. An additional external signal, denoted « timing 1/64 » in FIG.

15   2, is used to insert a reference pattern in the signal sequence as will be explained below and it is fed to both blocks serial/parallel converter and OQAM modulator.

In the system, at least one subchannel is used to carry a synchronization signal described hereafter and service data. The

20   corresponding signal xis is generated by the « synchro+data » unit **150**, the service data, denoted « scdatae » in FIG. 2, being supplied by the receiver **200** shown in FIG. 4.

The timing of the system is organized in 3 levels which will be referred to as follows.

25      1) **frame**:      it is the basic period in the system, associated with the subchannel spacing fs/N. For example, if fs/N = 4 kHz, a frame is 0.25 ms. The frame is used for payload transmission, amplitude subchannel equalization and

30                           synchronization.

2) **superframe**: a duration of N1 frames. For example, if N1 = 64 the superframe length is 16 ms. It is used for supervised subchannel equalization, noise level measurement and synchronization.

35      3) **hyperframe**: a duration of N2 superframes. For example, if N2 = 64 the hyperframe length is 1024 ms. It is used to confirm or modify the bit assignments in the subchannels.

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Turning to synchronization, at least one subchannel is used to transmit a specific signal. For example, it can be subchannel 69, whose central frequency is 4 kHz x 69 = 276 kHz. The specific synchronization signal is designed to provide an
5   efficient and robust control of the sampling times in the receiver and perform frame, superframe and hyperframe alignment. It contains the following superframe synchronization pattern.

SFP = { I I I -I -I -I -I I I I I -I -I -I -I I }

With the real and imaginary alternation required by the OQAM
10   technique, and the filtering operation performed by the filter banks, such a pattern produces, at the receiving end and in the absence of amplitude distortion in the transmission channel and no signal present in the neighboring subchannels, a complex signal whose squared magnitude v(n) is a fs/2N frequency sinewave
15   of amplitude 0.5, added to a zero-frequency component of amplitude 1.5, as shown in FIG. 3. With the numerical values given as example, fs/2N = 2 kHz, SFP has a duration of 2 ms and produces 4 periods of the 2 kHz sinewave, as is clearly apparent in the left part of FIG. 3. Since the superframe has a length
20   of 16 ms, 14 ms are available for the transmission of service data. These data are encoded as

P0 = ± { I -I -I I } for a « zero » ;   PI = ± { I I I I } for a « one »

Two consecutive data are separated by a period of the 2 kHz sinewave, as shown in FIG. 3.
25        The signal used to control the phase lock loop associated with the oscillator that delivers the received signal sampling frequency, or receiver clock generator, is obtained by filtering the 2 kHz component in v(n). This function is advantageously realized in two steps, as follows. A signal c(4n), with sampling
30   frequency 2 kHz, is obtained by

$$c(4n) = v(4n) - v(4n-I) - [ v(4n-2) - v(4n-3)] \qquad (3)$$

Then, an averaging operation is performed to attenuate the noise and the interferences from the neighboring subchannels

$$ca(4n) = (I-\varepsilon) \, ca[4(n-I)] + \varepsilon \, c(4n) \qquad (4)$$

35   where $\varepsilon$ is a small constant, for example $\varepsilon = 10^{-3}$. The signal ca(4n) is used to control the phase lock loop of the clock generator.

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The sign ± in P0 and P1 is used as shown in FIG. 3 to ensure that rising and falling edges of the 2 kHz signal keep a fixed relative position in time, regardless of the transmitted service data.

The hyperframe synchronization pattern HFP occurs every 64 superframes and it consists of a superframe in which the pattern SFP is repeated 4 times: HFP = {SFP,SFP,SFP,SFP}.

In view of equalization in the receiver, an additional feature of the OQAM modulation block **120** in the emitter, is that it imposes fixed values to the first two samples of the superframe in each subchannel, for example: ± [ 1; 1 ]. A specific sign may be attributed to each subchannel, in order to avoid producing a large peak in the emitted multicarrier signal Se(n) at the beginning of each superframe.

In the receiver, the multicarrier received signal Sr(n) is processed by a cascade of 4 blocks, namely the analysis filter bank **210**, a subchannel equalizer **220**, a data extraction module **230** and a parallel-to-serial converter **240**. The subchannel equalizer **220**, which consists of a cascade of 3 distinct equalizers, is shown in more details in FIG. 5.

Turning to FIG. 5, in subchannel i, the amplitude equalizer **221** receives the input sequence xir(n) and a reference amplitude ra supplied by the bit assignment module **250**. This reference amplitude ra represents the theoretical value of the root mean square of the power of the data signal in the subchannel. For 1-bit data, ra=1. The amplitude equalizer **221** computes a variable gain g1(n), multiplies the input signal by that gain and transfers the result yi(n) to the phase equalizer **222**. The variable gain is obtained through the two following operations

$$sgm(n+1) = (1-\varepsilon)\ sgm(n) + \varepsilon\ |xir(n+1)| \tag{5}$$

where $|x|$ stands for the modulus of $x$ and $\epsilon$ is a small real number, for example $\epsilon = 10^{-3}$.

$$g1(n+1) = ra\ /\ sgm(n+1) \tag{6}$$

Next, the phase equalizer **222** multiplies its complex input signal yi(n) = yir(n) + j yii(n) by a complex gain g2 = a + j b to produce the output ui(n).

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The gain is updated at the beginning of the superframe, using the first two samples, denoted yi(n) and yi(n+1). In fact, the following matrix system is solved in the least squares sense.

$$\begin{vmatrix} yir(n) & -yii(n) \\ yii(n+1) & yir(n+1) \end{vmatrix} \begin{vmatrix} a \\ b \end{vmatrix} = \begin{vmatrix} 1 \\ 1 \end{vmatrix} \tag{7}$$

5    A preferred approximate implementation of the least squares algorithm is as follows

$$a = A / C \quad ; \quad b = B / C \tag{8}$$

where the quantities A, B and C are updated every superframe by

$$A(p+1) = (1-\varepsilon) A(p) + \varepsilon [yir(n+1) + yii(n)] \tag{9}$$

10   $$B(p+1) = (1-\varepsilon) B(p) + \varepsilon [yir(n) - yii(n+1)]$$

$$C(p+1) = (1-\varepsilon) C(p) + \varepsilon [yir(n)yir(n+1) + yii(n)yii(n+1)]$$

The parameter $\varepsilon$ is a small constant, for example $\varepsilon = 10^{-2}$, p is the superframe index. The initial values can be A(0) = $10^{-3}$, B(0) = 0 and C(p) is kept no smaller than $10^{-2}$.

15   Once the complex gain elements have been calculated, a real error signal is derived as follows.

$$eip(n) = 1 - [a \, yir(n) - b \, yii(n)] \tag{10}$$

$$eip(n+1) = 1 - [b \, yir(n+1) + a \, yii(n+1)]$$

The error signal is used for noise level estimation as described

20   below.

The fine equalizer 223 computes the following output

$$vi(n+1) = ui(n) + \sum_{k = -1}^{1} h_k(n) \, ui(n-k) \tag{11}$$

25   to be delivered to the data extraction module 230. The function of the fine equalizer is to complete the task of the two previous modules, in particular to remove the residual distortion. Its coefficients $h_k(n)$ generally take very small values and they can be updated at the superframe rate, using the same reference

30   signal as the phase equalizer 222. In addition, they can be updated during regular transmission, according to the data directed equalizer principle, using the error signal ei(n) provided by the data extractor 230 and the least mean squares (LMS) algorithm

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$$h_k(n+1) = h_k(n) + \delta\, ei(n+1)\, ui(n-k) \tag{12}$$

where $\delta$, the adaptation step size, is a small value, for example $\delta = 10^{-3}$.

Turning back to FIG. 4, the capacity of a subchannel is determined by the « capacity+ bit assign » block **250**. This block receives the error signal eip from the phase equalizer and the error signal ei from the data extraction block **230**. It computes the following two variables.

$$E1(p) = (1-\varepsilon 1)\, E1(p-1) + \varepsilon 1\, [\; eip^2(n) + eip^2(n+1) \;] / 2 \tag{13}$$

$$E2(n) = (1-\varepsilon 2)\, E2(n-1) + \varepsilon 2\, [\; ei^2(n) \;] \tag{14}$$

where the parameters $\varepsilon 1$ and $\varepsilon 2$ are small values like $10^{-2}$ and $10^{-3}$ respectively for example.

The quantity E1(p) is computed every superframe and it is representative of the total distortion plus noise power present in the subchannel, before fine equalization. The quantity E2(n) is computed at the rate 8 kHz and it is representative of the noise power in the subchannel. In normal operation, with the above equations (13) and (14), E2(n) is smaller than E1(p) and the difference depends on the improvement brought by the fine equalizer.

Based on the results of these calculations, a decision is made at the beginning of each hyperframe to keep or modify the number of bits assigned to each subchannel. Then, the corresponding information data, denoted « scdatae », are fed to the « synchro+data » block **150** of the emitter **100**, for transmission to the distant terminal during the current hyperframe and to the data extraction block **230** in the receiver **200**, for use during the next hyperframe. The determination of the number of bits Nb assigned to the subchannel is a two-step process. First, Nb is calculated, for example through successive comparisons to thresholds, as

$$Nb = Int\, [\; \tfrac{1}{2}\, Log_2\, (\, 1/\, E1(p)) - 1 \;] \quad ; \quad E1(p) < 0.25 \tag{15}$$

where Int[x] stands for the integer part x. Then, E2(n) is used to confirm the decision or improve it. For example, if E2(n) is smaller than E1(p)/4, the number of bits may be increased by one.

The number of bits assigned to the subchannels is limited by the service data capacity. As pointed out earlier and shown

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in FIG. 3, the synchronization signal can transmit 14 bits of
service data in a superframe with the numerical values given.
If 3 bits are allotted to each subchannel, 4 subchannels can be
handled per superframe and, if 240 subchannels are actually used,
5 then, 60 superframes are sufficient to transmit the whole
capacity information. The number of bits assigned to any
subchannel is included in the range [0, 7].

The data extraction module **230** receives the signal $vi(n)$
from the subchannel equalizer **220** and performs a quantization
10 operation on the real and imaginary parts alternatively, using
the quantization scale associated with the number of bits
assigned to the subchannel. The binary representation of the
quantized value dir is fed to the parallel/serial converter **240**
and the quantization error $ei(n)$ is sent back to the subchannel
15 equalizer **220** to be used as per equation (12). The paral-
lel/serial converter **240** produces the output data stream $d'(n)$.

The «synchro processing » block **270** is shown in more details
in FIG. 6. It receives the synchronization subchannel signal
xisr and performs amplitude equalization through block **271**, as
20 described previously for the other subchannels. The signal
obtained, $yis(n)$ is fed to the « envelop detection » module **272**
that computes the variable $v(n) = |yis(n)|^2$. Then, the control
signal $ca(4n)$ is generated by block **273** as explained above and
according to equations (3) and (4).

25 The subchannel signal xisr is also fed to an amplitude/phase
equalizer **274** that produces a signal $uis(n)$, from which the
binary data at the rate 8 kHz are recovered, with the help of a
data detector **275**. In fact, the data detector just takes the
sign of the real and imaginary parts of $uis(n)$ alternatively.
30 The binary sequence bs(n) so obtained is fed to the « synchroni-
zation pattern and data extraction » block **276**, that recognizes
the superframe and hyperframe synchronization patterns and
delivers the corresponding timing information denoted « timing
1/64 » in the figures. The block also separates the bit assign-
35 ment data, denoted « scdatar » and delivered to the OQAM
modulator **120** of each subchannel and to the serial/parallel
converter **110** in the emitter of the system.

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Industrial Applicability

An important feature of the system of the invention is that
poor quality sections of the transmission channel frequency band
can be exploited, through the combination of several subchannels.
5    In each superframe, the synchronization subchannel signal carries
the bit assignment data for a group of 4 subchannels. If, for
these 4 subchannels, the noise power estimations Elj(p) with j=1,
2, 3 and 4, are all greater than 0.0625, which means Nb=0, and
if the following condition is satisfied

10
$$\sum_{j=1}^{4} \left( 1 / Elj(p) \right) > 16 \tag{16}$$

then, the same one-bit data signal is fed to these subchannels
in the emitter and the corresponding phase equalizer outputs
15   ui+j(n) in the receiver are summed as follows

$$sum(n) = \sum_{j=1}^{4} ui+j(n) / Elj(p) \tag{17}$$

and the input data are retrieved as the sign of the variable
20   sum(n). With that technique, a one-bit data sequence is
transmitted by 4 subchannels. Clearly, this is an example and
similar combinations can be elaborated for other numbers of
subchannels, like 2, 3, 8 or 16. The output vi(n) of the
subchannel equalizer 220 may also be used.

25       Although the present invention has been described in terms
of the presently preferred embodiment, it is to be understood
that such disclosure is purely illustrative and is not to be
interpreted as limiting. Consequently, without departing from
the spirit and scope of the invention, various alterations,
30   modifications, and/or alternative applications of the invention
will, no doubt, be suggested to those skilled in the art after
having read the preceding disclosure. Accordingly, it is intended
that the following claims be interpreted as encompassing all
alterations, modifications, or alternative applications as fall
35   within the true spirit and scope of the invention.

The Claims
            What is claimed is:


        1.    A multicarrier digital transmission system comprising
5  an emitter **(100)** and a receiver **(200)**:
            a.    the emitter **(100)** including:
                  i.    means **(110)** for splitting an input data stream
                  d(n) into a number of substreams equal to a number of
                  used subchannels;
10                ii.   means **(120)** for performing OQAM modulation on
                  each substream and producing a set of N sequences xi,
                  each sequence being coupled to the input of a synthe-
                  sis    filter    bank    means    **(130)**    that    produces    a
                  multicarrier output digital signal Se; and
15                iii.  means **(150)** for generating a synchronization and
                  service data signal xis in at least one subchannel of
                  the system; and
            b.    the receiver **(200)** including:
                  i.    analysis filter bank means **(210)** to produce from
20                an input signal Sr a set of sequences xir equal to the
                  number  of  used  subchannels,  each  sequence  being
                  coupled to an input of a subchannel equalizer means
                  **(220)**;
                  ii.   data extraction means **(230)** to process equalizer
25                output and feed a parallel-to-serial converter means
                  **(240)** that produces an output data stream d'(n);
                  iii.  means **(270)** for detecting a synchronization and
                  service data signal;
                  iv.   means **(250)** for estimating a noise level for
30                every subchannel and for deciding a number of bits to
                  be assigned thereto,
            said system being characterized by, a subchannel equalizer
        **(220)** in the receiver **(200)** that includes a cascade of amplitude
        equalization means **(221)**, phase and residual amplitude equaliza-
35  tion means **(222)** and fine equalization means **(223)**.


        2.    An emitter and a receiver as defined in claim 1 wherein
        the signal transmitted by the synchronization subchannel (s)

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contains a superframe synchronization pattern and a hyperframe synchronization pattern.

3. An emitter as defined in claim 1, wherein the synchronization and service data signal is transmitted by envelop modulation.

4. A receiver as defined in claim 1, wherein the envelop of the synchronization and service data signal is used to produce a signal to control the receiver sampling times and perform frame synchronization.

5. An emitter as defined in claim 1, wherein each superframe the OQAM modulator (120) introduces a short fixed sequence into the signal in each subchannel.

6. A subchannel equalizer (220) as defined in claim 1, wherein the phase and residual amplitude equalizer (222) is updated at a superframe rate, using a short fixed pattern introduced by the emitter as its reference signal.

7. A subchannel equalizer (220) as defined in claim 1, wherein the fine equalizer (223) is a decision directed equalizer.

8. An emitter and a receiver as defined in claim 1, wherein data for bit assignment in the subchannels are transmitted each hyperframe, and the numbers of bits carried by each subchannel are confirmed or modified each hyperframe.

9. An emitter and a receiver as defined in claim 1, wherein the same input signal is fed to two or more subchannels at the emitter and the outputs of the corresponding subchannel equalizers (220) are combined to produce a signal from which the data are extracted.

10. A combination of subchannel signals as defined in claim 9, wherein the combination is achieved by a weighted summation,

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the weighing coefficients being the inverses of the estimated total distortion powers, and the data are obtained by taking the sign of the weighted summation result.

5        11. An emitter and a receiver as defined in claim 1, wherein the coefficients in the filter banks (130) and (210) are such that the prototype filter response satisfies the first Nyquist criterion and the second Nyquist criterion.
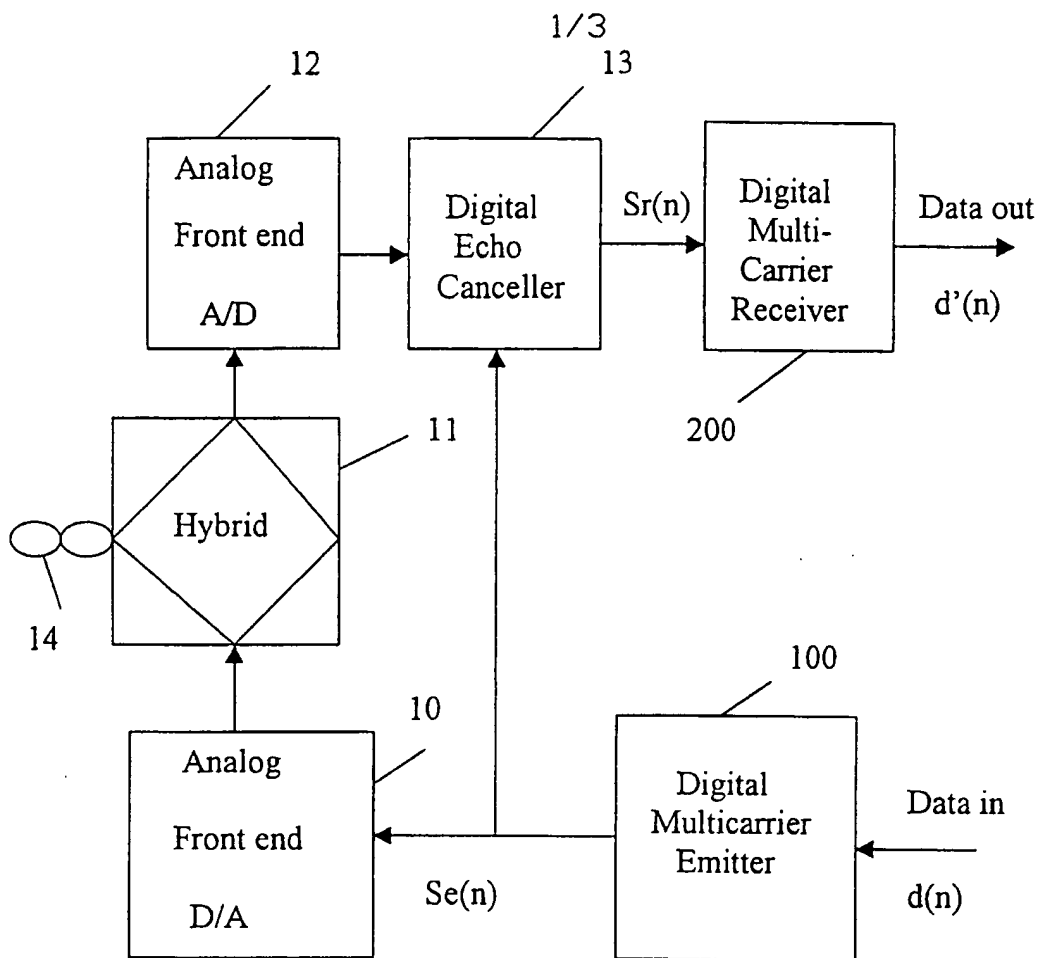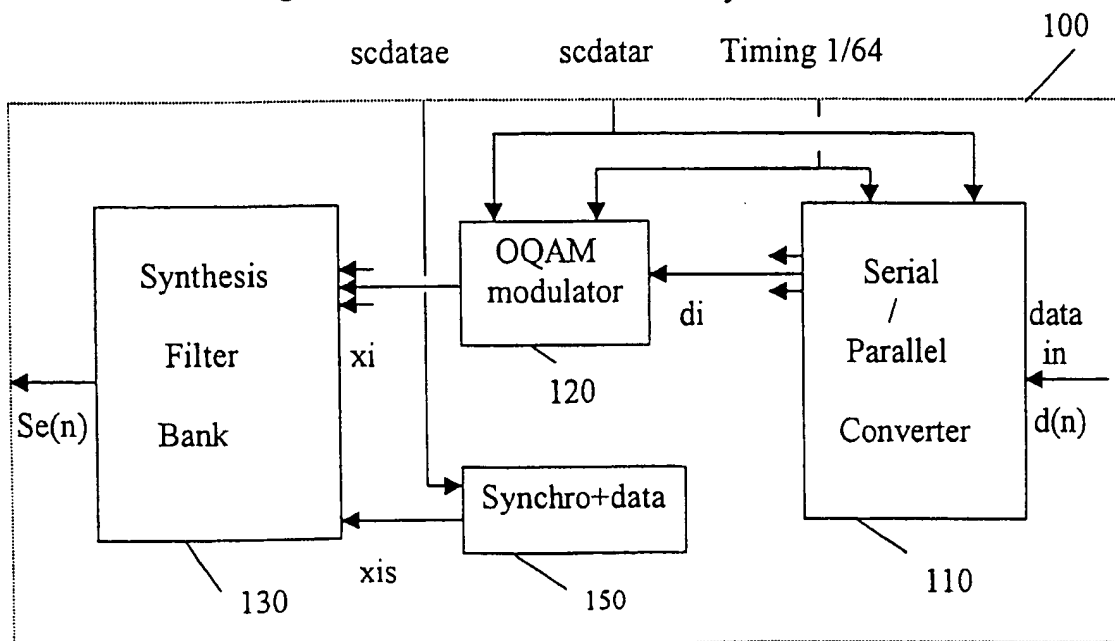
1/3



Fig.1. Multicarrier Transmission System



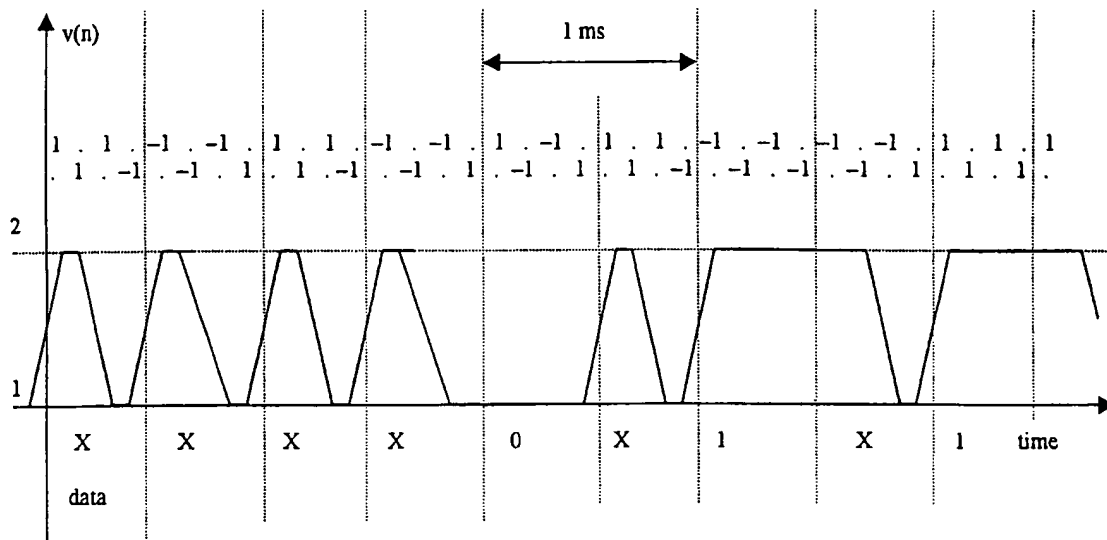Fig.2. Digital Multicarrier Emitter
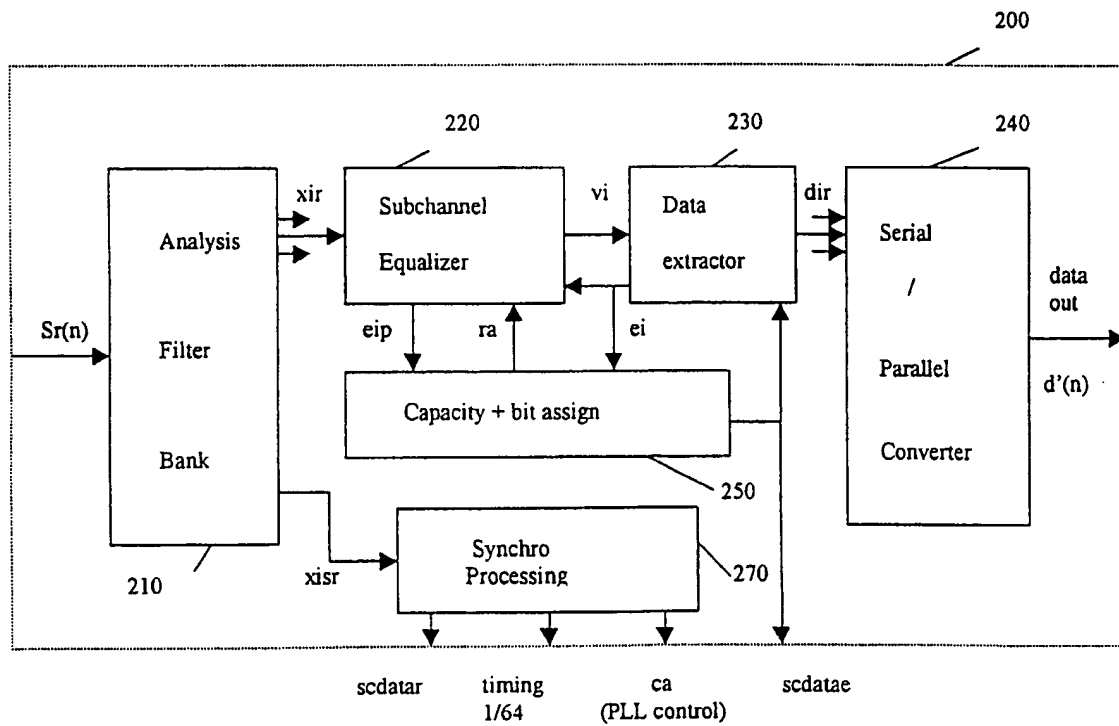
Fig.3. Synchronization subchannel signal



Fig.4. Digital Multicarrier Receiver

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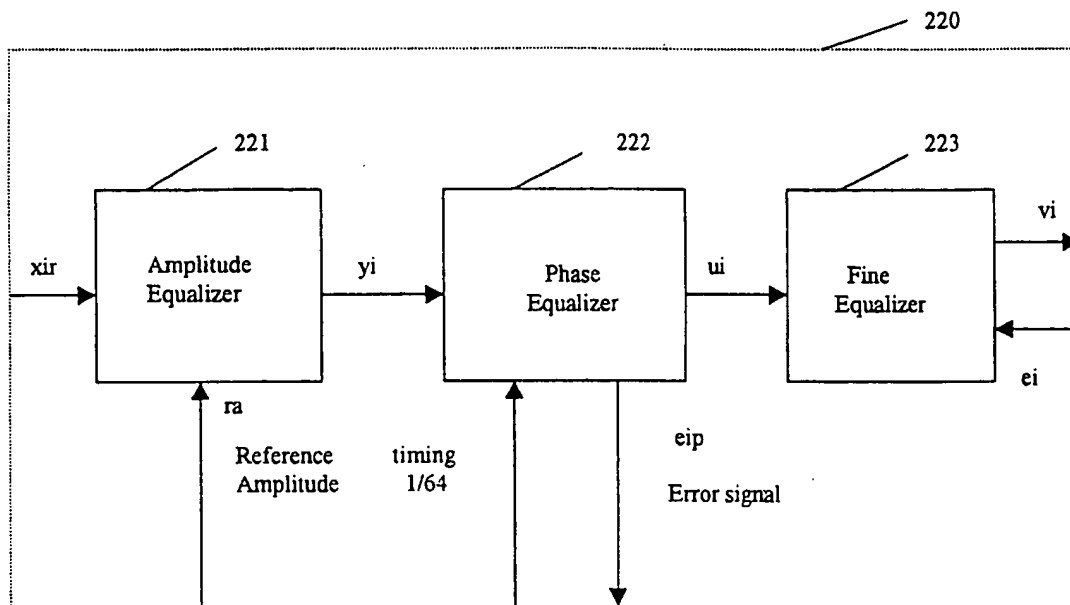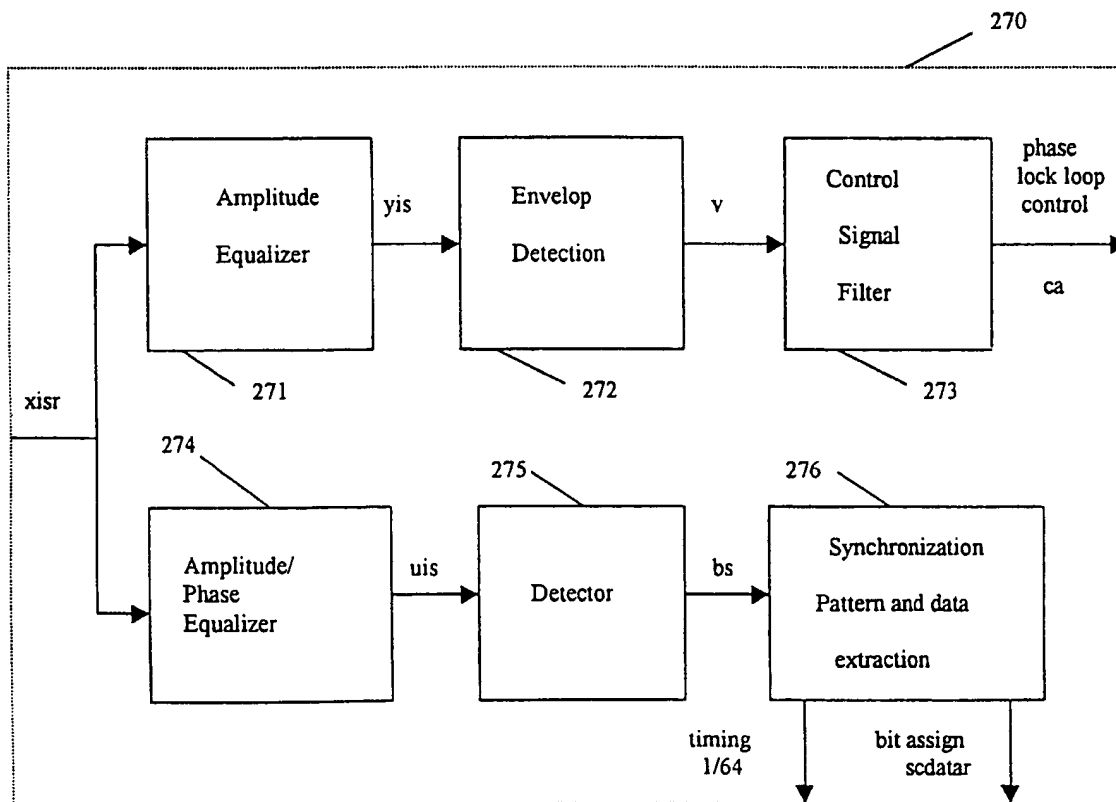Fig.5. Subchannel Equalizer



Fig.6. Processing of received synchronization channel

# INTERNATIONAL SEARCH REPORT

| International application No. |
|---|
| PCT/US00/42048 |

**A. CLASSIFICATION OF SUBJECT MATTER**

IPC(7) :H04J 1/20; H04L 7/00, 27/01, 27/34

US CL :375/230, 261, 298, 349, 350, 355; 370/483, 484, 497, 509

According to International Patent Classification (IPC) or to both national classification and IPC

**B. FIELDS SEARCHED**

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 375/229, 230, 232, 233, 260, 261, 298, 349, 350, 355; 370/480, 483, 484, 497, 509

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

NONE

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

EAST search terms: synthesis filter, analysis filter, equalizer, multicarrier, subchannel, synchronization, OQAM, frames, amplitude, phase, fine

**C. DOCUMENTS CONSIDERED TO BE RELEVANT**

| Category* | Citation of document, with indication, where appropriate, of the relevant passages | Relevant to claim No. |
|---|---|---|
| A, P | US 6,047,025 A (JOHNSON et al) 04 April 2000, see its entirety. | 1-11 |
| A | US 5,644,596 A (SIH) 01 July 1997, see its entirety. | 1-11 |
| A | US 4,853,802 A (KUKSON et al) 01 August 1989, see its entirety. | 1-11 |

☐ Further documents are listed in the continuation of Box C.   ☐ See patent family annex.

| * | Special categories of cited documents: |
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| Date of the actual completion of the international search | Date of mailing of the international search report |
|---|---|
| 06 FEBRUARY 2001 | 03 APR 2001 |

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| Commissioner of Patents and Trademarks<br>Box PCT<br>Washington, D.C. 20231 | AMANDA T. LE |
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